IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re the Application of	:	Docket No.	0805774-0004
POLETTI	:	Art Unit	2643
Application No. 09/197,096	:	Examiner	LAO, LUN S
Filed: November 20, 1998	:		

For: An Improved Guitar Preamplifier System With Controllable Distortion

DECLARATION

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I, MARK ALISTAIR POLETTI, declare:

1. I am the MARK ALISTAIR POLETTI named as inventor in US patent application 09/197,096 titled "An Improved Guitar Preamplifier System With Controllable Distortion".

2. I am employed as a research scientist at Industrial Research Limited in New Zealand which is a New Zealand Government owned research organisation. I hold a PhD (Acoustics) from the University of Auckland, New Zealand, 2001, and a MSc from the University of Auckland 1984, and BSc University of Auckland, 1982.

3. My areas of expertise include signal processing theory, digital signal processing, sound recording and reproduction systems, audio and acoustics systems. Attached marked MAP1 is a list of papers I have had published in peer reviewed journals and conference papers I have presented and I have also had two book chapters published.

4. In 2000 I received the New Zealand Royal Society award for the development of the VRA room acoustic enhancement system, which is licensed to the US company Level Control Systems (www.LCSaudio.com). In 1999 this system also won the Sound Product of the Year award, Lighting Dimensions International, Florida 1999.

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Maag – US Patent 589233

5. I have carefully read US patent 5,892,833 to Maag et al (herein Maag). This discloses a form of equaliser which produces an *overall* phase response which is claimed to be relatively constant across the audio frequency range. A stated goal of this system is to **reduce** distortion, which from the context of later statements (eg col. 2, line 58) clearly refers to linear amplitude and phase distortion rather than non-linear distortion.

6. An essential element of the guitar preamplifier system of my invention is a filtering means for splitting the input signal into multiple frequency bands which has a substantially equi-phase response for each frequency band. Splitting the input signal into frequency bands enables the desired harmonic distortion to be introduced in a controlled manner while reducing intermodulation products, which can introduce undesirable distortion that does not produce the pleasing musical effect. With equi-phase filtering, the multi-band distortion system produces a sound which is better again. The sum response has a flat frequency response, hence imparting no spectral colouration to the sound, and the distortion products generated in each band add in phase, preventing cancellation of the distortion products.

7. The result sounds more even, coherent, and natural relative to that of a multi-band system with non-equi-phase filters. Chords can be played with a reduction in intermodulation, which produces a harsh, buzzing sound, whereas with the invention, sound coherent and clear. For example, two note intervals sound better due to the reduction of intermodulation, and six note chords can be played with improved clarity. The output with zero-phase error is shown in Figure 3 of the patent application, and demonstrates no crossover-like artefacts. The result is also considerably better than standard solid state distortion circuits.

8. Specifically, the equi-phase response assists recombination of the frequency bands, after the desired non-linear distortion has been applied. The equi-phase response enables the desired sound to be retained when recombining the bands. To explain this further, with reference to a four band system, due to the finite selectivity of the four channels, a sinewave input with frequency f_0 will appear in all four channels, with a maximum amplitude in the band that is closest to f_0, and with reduced amplitude in the other three bands. The sinewave will be nonlinearly distorted in each band at high gains, producing harmonics at multiples of f_0, such as $3f_0$ and $5f_0$. Because of the equiphase channel responses, and the identical nonlinear

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distortion circuits, both the fundamental and all of the nonlinearly generated harmonics will have identical phases, and will therefore combine without cancellation occurring. This performance sounds subjectively superior to prior art systems with non-equal channel phase responses.

9. With a guitar input signal which comprises a sum of many sinewaves the behaviour of the circuit is more complicated. The equiphase bandsplitting means firstly that at low gain settings, the four bands recombine with a flat frequency response, eliminating spectral colouration of the guitar signal. Secondly, the equiphase bandsplitting means that each channel signal is a weighted sum of the original input sinusoids, and that these weighted harmonics retain the same phase relationship. The nonlinear distortion of the channel signals retains these equal phase relationships. Phase cancellation of distortion products is eliminated (or substantially eliminated).

10. The filters disclosed in Maag *do not individually produce substantially equi-phase responses.* It is only the sum response that is approximately constant phase. I have computer simulated and show below in Figure 1 the magnitude and phase responses of the output of operational amplifier 44 in Fig. 1 of Maag, with the filters in the filter bank defined in column 4 of Maag, with gains of one (36a to 36n equal to 52), and with center frequencies 10 Hz, 40 Hz, 160 Hz, 640 Hz, 2560 Hz and 10,240 Hz (ignoring the input attenuator which scales the magnitude and has no effect on phase, and ignoring the output filter formed by resistor 52 and capacitor 48, which – with the stated values of 11.1 kilohms and 22 picoFarads – has a cutoff frequency of 652 kiloHertz, well above the audio range):

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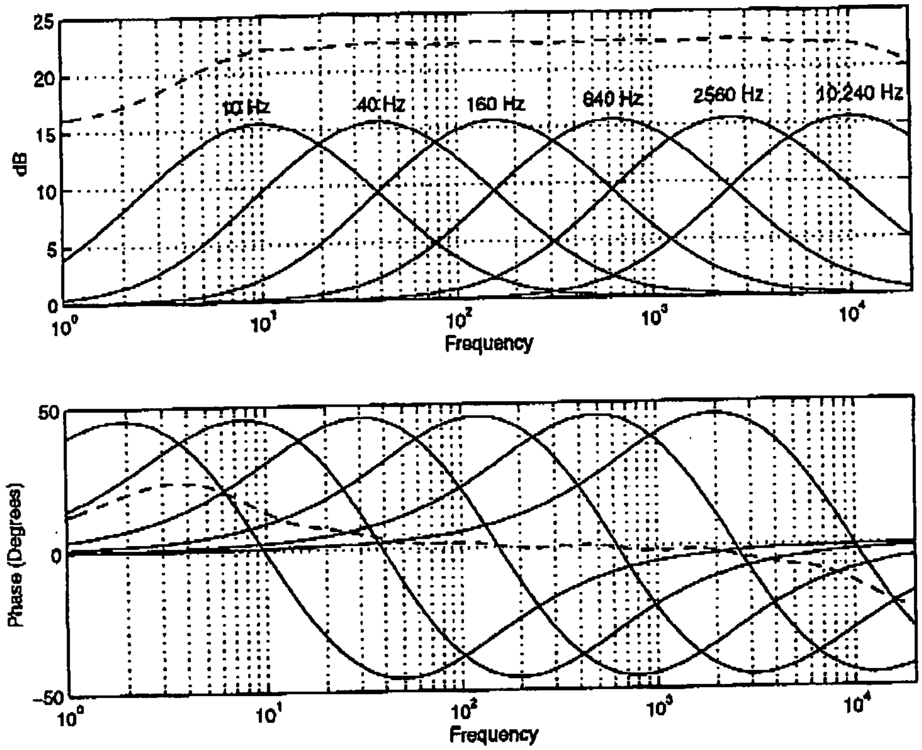

Figure 1: Magnitude (Upper) and Phase(lower) responses of the Maag filterbank with equal gains. The sum responses is shown dashed

11. As can be seen from the above figure, the phase responses in each channel are markedly different at any given frequency: for example, at the frequency 160 Hz the phases are approximately 45, 25, 12, 7, -35 and –41 degrees, a variation of 86 degrees total. The magnitudes sum to a flat response (with a small ripple of about 0.3 dB) as claimed, and the overall phase response which is shown by the dashed line has a relatively low variation across frequencies, with peaks at low and high frequencies, as claimed.

12. The guitar preamplifier of my invention requires equal phases *per channel*. The phase responses of each channel substantially do not change with the channel gains, which occur after the equaliser filters. However, the combined output phase does vary with the gains. For example, with gains 1,0,1,0,1,0, the response in figure 2 below is produced. The individual channel phases are the same, but the combined phase varies more extremely with frequency:

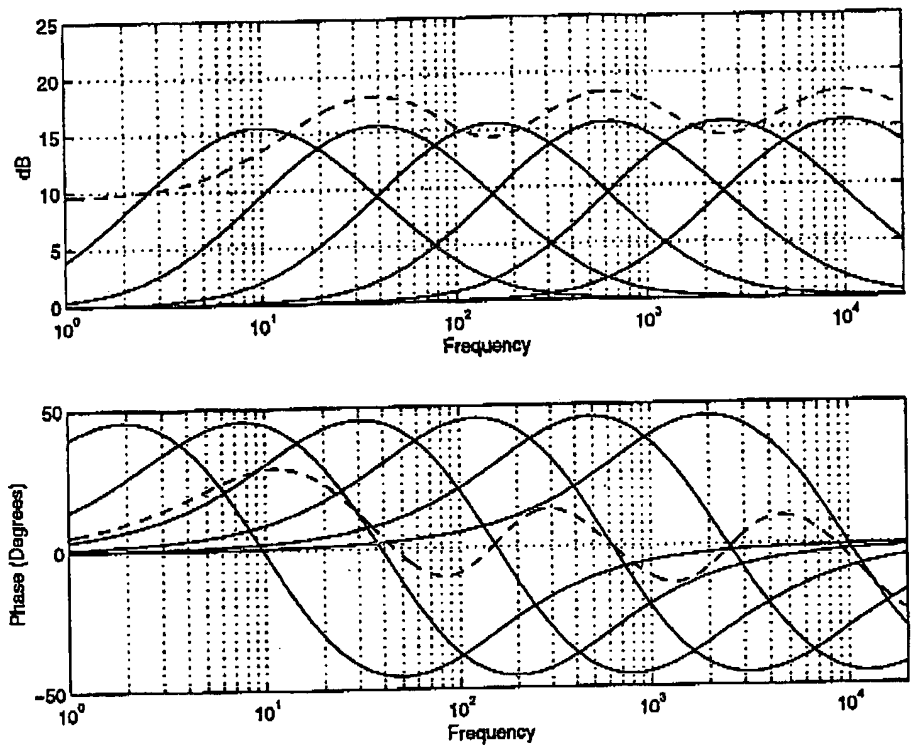

Figure 2: Magnitude (Upper) and Phase(lower) responses of the Maag filterbank with gains 1,0,1,0,1,0. The sum responses are shown dashed

Levine – US Patent 5,848,164

13. I have also carefully read US patent 5,848,164 to Levine (herein Levine). The objective of an electric guitar preamplifier is to deliberately introduce harmonic distortion through non-linearity. Levine does not teach this but teaches only linear circuits. Levine does not relate to guitar preamplifiers, but to subband processing to reduce the processing cost of implementing linear, time-varying digital effects such as chorusing, flanging, and reverberation. Levine discloses a linear, time-varying system not a non-linear effects system. Chorusing and flanging are both based on delaying a signal and adding the delayed version to the original, undelayed signal. This produces a series of peaks and dips in the frequency response where reinforcement and cancellation occur due to the linear phase shift produced by the delay. The chours and flange effects occur when the delay is changed with time, and produce a sweeping, shimmering sound effect. Flanging tends to use short delays which produces widely-spaced notches in frequency and a more extreme sound effect, whereas chorusing tends to use longer delays, with more closely-spaced notches in frequency, and this more closely simulates having several instruments playing instead of

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one. Both chorus and flange effects can be accentuated by feeding back some of the output signal to the input. Reverberation consists of generating reproductions of the input signal (echoes) of increasing density over time. The effect is again produced using multiple delays with feedback around them but the delays are longer than in chorusing, and the delays are not required to be altered to produce the reverberation effect. The echoes are not distorted. These are all linear effects, and not non-linear distortion effects as are intentionally generated in an electric guitar preamplifier. With a guitar preamplifier the amplifier intentionally produces a direct distortion of the amplitude of the input signal, through non-linearity.

14. I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements are made with knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of this Application for Patent or any patent issuing thereon.

DECLARED at Industrial Research Ltd)
Lower Hutt, New Zealand)
this 25th day of February 2004)

Mark Poletti
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MARK ALISTAIR POLETTI

"MAP1"

Mark Poletti - Publications

Publications List

Book Chapters

1) M. A. Poletti, "Linearly swept frequency measurements and the Wigner-Ville distribution," in *Time-Frequency Analysis: Methods and Applications*, Editor B. Boashash, Longman Cheshire/ Wiley Halsted Press, 1992
2) M. A. Poletti, "Assisted reverberation systems for the control of auditorium acoustics," Current Topics in Acoustical Research, vol. 1, pp 501-513, 1994, published by the Council of Scientific Information, India

Papers

1) M. A. Poletti, "Linearly swept frequency measurements, time delay spectrometry and the Wigner distribution," *J. Audio Eng. Soc.*, vol. 36, no. 6, pp 457-468, June 1988
2) M. A. Poletti, "The development of a discrete transform for the Wigner distribution and ambiguity function," *J. Acoust. Soc. Am.*, vol. 84, no. 1, pp 238-252, July 1988
3) M. A. Poletti, "The application of linearly swept frequency measurements," *J. Acoust. Soc. Am.*, vol. 84, no. 2, pp 599-610, August 1988
4) M. A. Poletti, "Instantaneous frequency and conditional moments in the time-frequency plane," *IEEE Trans Signal Processing*, vol. 39., no. 3, pp 755-756, March 1991
5) M. A. Poletti, "The development of instantaneous bandwidth via local signal expansion," *Signal Processing*, vol 31 no. 3, pp 273-281, 1993
6) M. A. Poletti, "On controlling the apparent absorption and volume in assisted reverberation systems," *Acustica*, vol 78, pp 61-73, 1993
7) M. A. Poletti, "The performance of a new assisted reverberation system," *Acta Acustica*, 2 December 1994, pp 511-524
8) A.J. Coulson, R.G. Vaughan and M.A. Poletti, "Frequency Shifting using Bandpass Sampling," *IEEE Trans. Signal Processing,* vol. 42, no. 6, pp 1556-1559, 1994
9) M. A. Poletti, "The design of encoding functions for stereophonic and polyphonic sound systems," *Journal of the Audio Engineering Society*, vol. 44, no. 11, pp 948-963, November 1996
10) M. A. Poletti, "The analysis of a general assisted reverberation system," *Acta Acustica* Vol. 84, pp 766-775, July/August 1998
11) M. A. Poletti, "The Homomorphic Analytic Signal," *IEEE Trans. Signal Processing*, vol. 45, No. 8, pp 1943-1953, August 1997
12) M. A. Poletti, "The statistics of single channel electroacoustic systems," *Acta Acustica* Vol. 84, pp 1077-1082, November/December 1998
13) M. A. Poletti, "The stability of single and multichannel sound systems," Acustica-Acta Acustica, vol. 86, pp 123-178, 2000
14) M. A. Poletti, "A Unified Theory of Horizontal Holographic Sound Systems," Journal of the Audio Engineering Society, vol. 48, no. 12, pp 1155-1182, December 2000
15) M. A. Poletti, "Direct and reverberant power analysis of multichannel systems," Acta Acustica, vol. 87, pp 531-541, 2001

Conference Papers

1) M. A. Poletti, "Frequency analysis of linear fm signals," delivered at the 8th Acoustics Conference, Auckland, July 1985
2) M. A. Poletti, "Elimination of the sweep rate limitation in linearly swept frequency measurements," delivered by G. Dodd at the 9th Acoustics Conference, Wellington, August 1987

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3) M. A. Poletti and R. G. Vaughan, "Reduction of multipath fading effects in single variable modulations," 2nd IASTED International symposium on signal processing and its applications, Gold Coast, Australia, pp 672-676, August 1990,
4) M. A. Poletti and R. G. Vaughan, "Reduction of multipath fading effects in single variable modulations," proc. 27th National Electronics convention, pp 251-262, September 1990,
5) M. A. Poletti, "Application of the Wigner-Ville distribution to time delay spectrometry," 2nd IASTED International symposium on signal processing and its applications, Gold Coast, Australia, pp 900-903, August 1990
6) M. A. Poletti, "The homomorphic analytic signal," 3rd IASTED International symposium on signal processing and its applications, Gold Coast, Australia, pp 49-52, August 1992
7) A.J. Coulson, R.G. Vaughan and M.A. Poletti, "Interpolation in bandpass sampling," 3rd IASTED International symposium on signal processing and its applications, Gold Coast, Australia, pp 23-26, August 1992
8) M. A. Poletti, "An Improved Assisted Reverberation System," Proceedings of the 12th Biennial Conference of the New Zealand Acoustical Society, pp 107-115, August 1993
9) M. A. Poletti, "Colouration in Assisted Reverberation Systems," Proceedings of ICASSP, 1994, Adelaide, Australia, Vol. 2, p 269-272
10) M. A. Poletti, "A unitary reverberator for reduced colouration in assisted reverberation systems," Proceedings of the 13th Biennial Conference of the New Zealand Acoustical Society, 1995
11) M. A. Poletti, "A unitary reverberator for reduced colouration in assisted reverberation systems," ACTIVE95, Newport Beach, California, July 6-8 1995
12) M. A. Poletti, "An assisted reverberation system for controlling apparent room absorption and volume," 101st convention of the Audio Engineering Society, Los Angeles, November 8-11, 1996
13) M. A. Poletti, "Assessment of colouration in assisted reverberation systems," Proceedings of the 14th Biennial Conference of the New Zealand Acoustical Society, 1997, p 5-2.1 - 5-2.10
14) M. A. Poletti, "A measurement of colouration in electroacoustic enhancement systems, Proc 16th International Congress on Acoustics/135th meeting of the Acoustical Society of America, June 20-26 Seattle, 1998
15) Y. Cao and M. A. Poletti, "Advanced system identification techniques for acoustic enhancement," Proc 16th International Congress on Acoustics/135th meeting of the Acoustical Society of America, June 20-26 Seattle, 1998
16) M. A. Poletti, "A comparison of passive and active coupled rooms for acoustic control" Internoise 98, Christchurch 16-18 November
17) M. A. Poletti, "Equalisation of assisted reverberation systems," Sixth International Congress on Sound and Vibration, Copenhagen, July 1999
18) M. A. Poletti, "The philosophy of electronic room enhancement," 139th meeting of the Acoustical Society of America, Atlanta, 30 May – 3 June 2000
19) J. Morris and M. Poletti, "Design and implementation of a multi-channel digital audio interface," Proceedings of the seventh Electronics New Zealand Comference, August 2000 (ENZCon '00).
20) M. A. Poletti, "The philosophy of the Variable Room Acoustics System," 15th Biennial Conference of the New Zealand Acoustical Society, 7-8 September 2000
21) S. Ellison and M. Poletti, "Variable Room Acoustics System: Philosophy and Applications", Institute of Acoustics, Proceedings, Volume 22 Pt 6 2000, p 215-223
22) M. A. Poletti, "The Equalisation of a Microphone Array for Surround Sound Recording," International Congress on Acoustics, Rome 2001

Other Professional Activities

- Member of Audio Engineering Society
- Member of IEEE
- Reviewer for IEEE Transactions on Signal Processing (9 papers reviewed)

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